WHAT IS CLAIMED IS:

1. A voice recognition system comprising:

an A/D converter that converts analog voice signals to digital signals;

an FIR filtering section that employs powers-of-two conversion to filter the 12-bit digital signals converted at said A/D converter into prescribed numbers of channels;

a characteristic extraction section that extracts voice characteristics having strong noise-resistance from the output signals of said FIR filtering section;

a word boundary detection section that discriminates the information of the start-point and the end-point of voice signal on the basis of the characteristics extracted by said characteristic extraction section; and

a normalization/recognition section that codes and outputs the final result by carrying out a timing normalization and a classifying process using a radial basis function(RBF) neural network on the basis of the voice characteristics provided by said characteristic extraction section and the information of the start-point and the end-point of voice signal from said word boundary detection section.

2. A voice recognition system as claimed in claim 1, wherein said characteristic extraction section is characterized by directly calculating the characteristic vectors at zero-crossing point of FIR filter output and accumulating them without storing the output of FIR filtering section.

3. A voice recognition system as claimed in claim 2 further comprising registers for each channel to accumulate said calculated characteristic vectors.

4. A voice recognition system as claimed in claim 3, wherein said registers comprise:

a register for accumulating the characteristic vectors between the total time interval(110 samples);

registers for accumulating the characteristic vectors only for the valid time of each channel; and

a buffering register for storing the characteristic vectors of the total time interval(110 samples).

5. A voice recognition system as claimed in claim 1, wherein said FIR filter is a cochlea FIR filter having limited coefficients.

6. A voice recognition system as claimed in claim 5, wherein said FIR filter is characterized by containing a command language to limit the coefficients of powers-of-two conversion by using the characteristics of said cochlea FIR filter.

7. A voice recognition system as claimed in claim 1, wherein said FIR filter is characterized by embodying a filter-bank with only additions and shift-operations by using powers-of-two conversion.

8. A voice recognition system as claimed in claim 1 further comprising a non-synchronized SRAM,

wherein said SRAM is characterized by storing the characteristics extracted from said characteristic extraction section and being read by said normalization/recognition section.